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Delay Monitoring in ATM Networks

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DELAY MONITORING IN ATM NETWORKS *

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ABSTRACT

ATM networks provide end-to-end QoS guarantees to connections for their lifetime, in the form of bounds on delays, errors and losses. Performance management involves measurement of these parameters accurately and taking control measures if required, to improve performance. This is very important for real time connections in which losses are irrecoverable and delays cause interruptions in service. In this paper, we concentrate on delay monitoring mechanisms. After presenting the OAM standard for ATM and a few solutions in the literature, problems still remaining are formulated and directions being pursued to obtain solutions are indicated.

1 INTRODUCTION

Asynchronous Transfer Mode (ATM) is a connection oriented fast packet switching protocol. Information is carried in fixed size cells along virtual channels, which are contained in virtual paths. ATM is designed to support various kinds of connections, voice/video/data etc. using statistical multiplexing of sources to achieve efficient resource utilization.

Every incoming call is characterized by its type, Constant/Variable/Available/Unspecified Bit Rate, and the Quality of Service it requires. The QoS of a connection consists of parameters like Cell Transfer Delay (CTD), Cell Delay Variance (CDV), Cell Loss Ratio (CLR), Cell Error Ratio (CER) etc. These parameters are important for real time connections (for example,

video-conferencing applications) in which delays are interruptions in service and losses may not be recoverable using retransmissions. Real time connections are usually CBR or VBR depending on the coding of the source.

In the connection setup phase a route is determined, with the network guaranteeing (the guarantee is of a statistical nature) the QoS required by the connection along the complete path. The network is obligated to provide this QoS throughout the duration of the connection but is allowed to drop cells if the source violates the QoS contract. To this end, performance management i.e., measurement and control of the QoS being provided to a connection, is necessary for the network. It is also necessary for congestion control, traffic management, planning in the network and possibly for billing. Measurement of QoS parameters for a channel (or path) is useful for the connection user or service provider also to know if the agreed QoS is being provided.

Performance management encompasses gathering of network performance statistics (in the form of QoS parameters), analyzing its trends and taking proactive or reactive control measures. In this paper we concentrate on the delay monitoring aspect of performance management. The next section presents the related standard and earlier work in this area. In subsequent sections various aspects of the problem are studied. The main purpose of the paper is to formulate the problems and indicate the directions of research being taken to approach the solution. Conclusions are presented in the last section.

2 STANDARD AND RELATED WORK

2.1 The Operations Administration and Maintenance (OAM) Standard for ATM

The OAM standard for ATM is recommendation I.610 from ITU-T [1]. Performance monitoring is done by inserting end-to-end or segment monitoring OAM cells at

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F4 (VP) or F5 (VC) level. These cells monitor a block of user cells (of size 128, 256, 512 or 1024). The main objective of this monitoring is to detect errored blocks and loss/mis-insertion of cells. There are forward monitoring cells to measure and backward reporting cells to report the measured values to the source. The various fields defined for these cells include a sequence number, the total number of user cells sent (a modulo 64k counter) and the number of user cells with high Cell Loss Priority (CLP = 1), a block error detection code (even parity/Bit Interleaved Parity - 16), and an optional time stamp (for delay measurements). For backward monitoring cells, the fields are a sequence number, total number of user cells received, number of user cells with CLP=1, and the block error result on the received cells.

Using these fields, block errors and differences in the number of transmitted and received cells can be determined. However, this does not give precise information about lost and mis-inserted cells. Only round trip delays are accurately measurable using the optional timestamp field (which is not implemented by most vendors currently) and that does not provide information about one-way delay, since the clocks of the source and destination usually are not synchronized, or bottle-neck links on the circuit.

It is noted that this provides limited information regarding the delays and losses which are critical to real time services. The standard does not specify how these values are used for proactive or reactive control actions, required if the QoS guarantees are not being met. It also does not specify how frequently these measurements should be made; this will depend on the bandwidth of the path, the network state, and user requirements for QoS. The next sections are devoted to review of existing papers on delay measurement [2].

2.2 Delay measurement using management cells

The one-way cell delay can be accurately measured by breaking it into delays experienced at each switch [3]. A switch can time-stamp a management cell when it is received at the input. At the departure of the cell, the cell delay can be computed and added to the delay value already written in the cell (initialized to zero by the source). The processing delay required for this scheme and the propagation delays are fixed and can be precomputed. Thus, a management cell accumulates the delay along the path. The delay field at the destination gives a sample of the cell transfer delay which does

not suffer from the clock synchronization problem. An alternative is to have multiple time-stamp fields in the management cell so that the delay at each switch can be recorded separately. This may be useful for diagnostic purposes, for example to determine the bottleneck link.

Management cells are required to occupy a small portion of the bandwidth, especially when the network is congested. This criterion dictates the inter-sample time. The authors of [3] have briefly alluded to this issue and stated some heuristic arguments. This scheme requires new processing capabilities at the switches to modify cells on ingress and egress. No mention is made about the accuracy of such a scheme either (which can possibly justify the additional complexity in the switches). The next section describes a procedure based on statistical analysis of the remote clock which can be used with OAM performance management cells.

2.3 Delay measurement using clock parameter estimation

The technique proposed by Roppel in [4] relies on estimation of the one-way cell transfer delay by analyzing the properties of the remote clock. Essentially, the remote clock can be modeled, its parameters estimated, and the time-stamp of the destination can be corrected for the offset. In this method, the switches do not need any new processing capabilities.

The remote clock can be modeled using a time offset parameter (ΔT_o) and a clock frequency drift parameter (α) $C(t) = t + \Delta T_o + \alpha(t - t_o) + \epsilon(t)$. These two parameters can be estimated using a regression model on the delay samples from the backward reporting performance management cells. This method however requires that the minimum delay along both directions to be the same, so it is not suitable for asymmetric and hybrid networks. The number of samples required to converge to correct clock parameters may be large, thus the time for convergence for low bit rate links can be very high. These disadvantages can prohibit the use of this scheme for a large class of networks and connections.

The above scheme can however be modified so that each node in the network estimates the parameters of its neighbors' clocks. This can be done with better accuracy as this involves only one queueing delay in the path, and may be done more simply and accurately using a CBR connection during light loading. Once a node knows the equations of all neighbor clocks, remote clock offset from source can be known by propagating

it hop-by-hop along the connection path at the connection setup time. Thus, the source can know the model for the remote clock at connection setup time and use it to correct timestamp values in OAM cells from the destination. This allows for measurement of one-way delay, although bottle-neck links can not be known. This scheme, which uses the performance management OAM cell, neither involves a large time delay to achieve accuracy nor new processing capabilities at switches and can be more accurate than Roppel's scheme.

3 ACCURATE DELAY MEASUREMENT SCHEMES

The OAM performance monitoring cell suggested in the ITU-T I.610 standard [1] is capable of measuring the round-trip delay only. Our aim here is to measure the one-way delay accurately, and try to identify the bottlenecks in a real-time circuit.

3.1 Loopback based

One of the simplest schemes that can be used for this purpose is *back flooding*. The OAM performance management cell is looped back by all the switches in the path, giving the source an estimate of the round trip delays to each node in the path (figure 1).

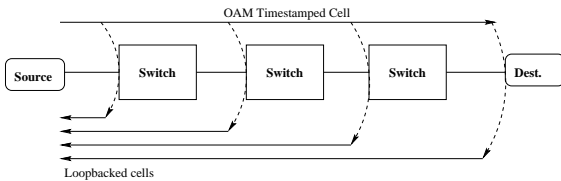


Figure 1: **Back - Flooding of OAM performance management cells.**

However, this results in very high overhead, measures a lot of unnecessary parameters and still suffers from the problem of measuring round-trip delays. An advantage of this scheme is that it is similar to the host-loopback, which is a kind of connectivity check using loopback cells, and thus should be easily implementable. It can be modified in a way that looks for the bottleneck nodes, by including in each cell a delay parameter value. Loopbacks at a node are performed only if the queueing delay at that node is more than the parameter. The looped back cell carries the address of the bottleneck switch. This new scheme termed *selective back-flooding* is better in that it has a smaller bandwidth overhead associated with it.

3.2 Explicit delay measurement based

The disadvantage of both the schemes described earlier is that the measurement is still of the round trip time which may not be very useful in some situations. To get around this problem, measurement of inter-departure times of consecutive OAM cells is proposed (*explicit delay measurement*). The source measures the inter-departure time of two consecutive performance management OAM cells of a new type, say *explicit performance measurement* type (can be specified using the free bits in the cell). A switch on the path also measures the same statistic for this type of cell and reports it back to the source (figure 2). Using the reports from all the switches in the path, the source can pinpoint the bottleneck nodes.

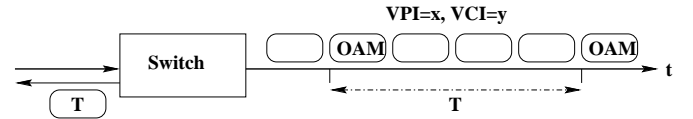


Figure 2: **Explicit measurement of delay.**

The advantages here are that each measurement is a difference of times on the same clock (and thus has no clock synchronization problem), and the one-way delay can be known here. The disadvantages of this scheme are that new processing capabilities are required at the switches which may not be easily implementable, and the bandwidth overhead of the scheme is high. Also, the inter-departure time measurement should be easier to implement than the queueing delay measurement suggested by [3]. It is suggested that this scheme be used as a diagnostic, in conjunction with selective back-flooding or even OAM performance management with timestamped cells.

3.3 Information Push schemes

Another method to pinpoint a bottleneck is using *soft-alarms*. The inter-departure times of real-time traffic queues can be measured and alarms (called soft-alarms because they indicate soft faults) can be sent back to the source of the VC when the delay exceeds a threshold (which can be agreed upon at connection setup). Alarms can also be sent out if the VC queue size exceeds a threshold (note that these two criteria represent different information). This is the information-push paradigm as opposed to all the above schemes which are based on the information-pull paradigm. The bandwidth overhead here is much smaller than all the schemes above for obvious reasons.

3.4 Future work

There still remains a need for schemes which have minimal bandwidth overhead, are able to extract all important bottleneck information from the network, are capable of accurately measuring various queueing and propagation delays in the network, and have minimal implementation complexity. It can be expected that the optimal scheme would be a combination of push and pull algorithms and would be dependent on the class and QoS of the traffic. Efficient schemes with both paradigms have to be found and optimal mixtures for various types of traffic have to be defined. The optimality would be defined using the notions of accuracy and overhead, as discussed more later.

4 THE SAMPLING FREQUENCY

The bandwidth overhead associated with measurements varies linearly with the frequency of measurements. To complement an efficient and accurate measurement algorithm, an optimal sampling policy has to be defined. The aim now is to send cells at a rate that captures any major change in the delay and is also the minimum such rate. In the ITU-T standard [1], it is proposed to monitor a block of cells of average size 128/256/512 or 1024. This method is bandwidth inefficient as it has a constant overhead.

An algorithm with characteristics: start sampling at a very high rate, reduce rate when enough information regarding the pattern is available and increase the rate as required by changes in the pattern and bring it down as soon as possible, is an ideal candidate for such an application. This is a conservative algorithm, achieving the complete opposite of a greedy algorithm. The parameters of this algorithm are: the initial rate and the criterion used to decide the increase and decrease of the sampling rate. It is proposed to use the initial rate as the maximal admissible rate $R_{max}(t)$ allowed by the maximum admissible measurement overhead that can be sustained on the network. This is a function of time as it is a fraction of the data bandwidth which may be variable (for example in VBR sources). An algorithm of this class is used for polling in a network management application in [5].

We now propose a new algorithm for determining the optimal sampling policy. The increase and decrease of the sampling rate is proposed to be on the basis of the Nyquist sampling frequency of the spectrum of samples in a window of size N . The input to the algorithm here

is a sequence of delay samples and their corresponding times $d(t) = \sum_{i=0}^N d(t_i) \delta(t - t_i)$. Note that this is not a discrete time signal and has timing information associated with it. The algorithm now is:

1. At time t_0 (marked by the arrival of a delay sample) convert the samples in the window of size N to an analog signal using a Low Pass Filter with bandwidth B (the maximum sampling frequency used in that window).
2. Using the spectrum of the signal, the new optimal sampling frequency ($F_{samp}(t_0)$) can be found by evaluating the Nyquist frequency of the signal if it is band-limited by B . If the signal has a bandwidth near the LPF bandwidth, $F_{samp}(t_0)$ should be much higher than $2B$. The sampling rate is always upper limited by $R_{max}(t)$, i.e., $F_{samp}(t) \leq R_{max}(t)$.
3. The next delay monitoring cell is sent out at time $t_0 + F_{samp}(t_0)^{-1}$.
4. At time $t_1 \geq t_0$, the time of arrival of the next monitoring cell, advance the window by 1 and go back to step 1.

Our future research in this direction includes a better formulation of the algorithm (including a specification of $R_{max}(t)$ and N), a theoretical verification of the correctness and optimality of the algorithm, and a practical demonstration of such a scheme using a real ATM switch and a traffic generator.

5 MEASUREMENT THEORY FORMULATION

In addition to schemes for accurate measurement and algorithms for optimal sampling of delay patterns, there is a need for a formalism to understand these algorithms and to evaluate their performance. Since these are measurement based schemes for finding the delay pattern as a function of time rather than the average over an interval, it is not desirable to use queueing theory here which is probabilistic and useful for obtaining statistics of the measurements.

Our aim here is to derive a theory in which the concepts of accurate measurement of delay (including measurement of delay at bottleneck points), and the notion of optimal sampling frequency are naturally embedded. It should naturally define the bandwidth overhead for measurements and the accuracy of measurements. It is

expected that precise definitions of these two quantities would give a tradeoff equation between them.

To this end, Flow Theory [6] is used with many modifications. The theory as presented in [6] makes use of the definition of a flow as a infinite real valued sequence r_0, r_1, \dots, r_i representing the number of packets in flow at instant i . The notions of (m, R) smoothness and uniformity are defined and several theorems regarding a buffer's capacity and delay under operators like limiter, compactor and expander are proven [6]. Theorems for cascade of operators are given and application of this theory to analyze rate-reservation protocols is given. Using these results, one is able to obtain bounds on the buffer capacity and delay, thereby helping design of a cascade of buffers [6]. Since the theory is not probabilistic, bounds are obtained and not statistics of the parameters.

We are currently investigating use of a variant of this theory to analyze the time evolution of queueing delay in the network. This requires representing the flow in continuous time rather than in discrete time. The representation now includes timing information associated with the flow also. Let the input to the network be only a stream of timestamped performance management cells which are looped back by the destination. Assume that the one way delay to the destination can be known. The incoming cells provide us with delay as a function of time $Delay(t) = \sum_{i=0}^N d(t_i) \delta(t - t_i)$. Now multiplex this stream with a data stream, that does not loop back (for example video data). The new delay time function is of the form $Sample(t) = \sum_{i=0}^N \hat{d}(t_i) \delta(t - t_i)$ where $\hat{d}(t_i) = d(t_i)$ for $i = j_0, j_1, \dots, j_m$ and zero otherwise.

The important step now is to define the bandwidth overhead in the Sample flow and the accuracy of the flow which would be the distance between the two functions $Delay(t)$ and $\hat{Sample}(t)$. The function $\hat{Sample}(t)$ is obtained from $Sample(t)$ by extrapolating it, i.e., passing it through a Low Pass Filter of bandwidth B , which defines the maximum sampling frequency used thus far. The definition of accuracy also has to include any extra information available, for example the congestion points in that path etc.

Our future research includes formalizing these definitions, defining conformance criteria on them (which represent loss/gain of information content in the sampled stream with respect to the actual stream), obtaining results on what happens to the streams and their conformance as they pass through various operators (representing switches) - for example buffer overflows,

using these results to define the distance between a real and sampled stream and using the definitions to obtain the tradeoff equations. It is expected that such tradeoff equations would prove instrumental in deciding the monitoring policy to be chosen for a particular class and QoS.

6 CONCLUSIONS ¹

Performance management in ATM involves measurement of the QoS parameters of a class of service. In this paper, the problem of delay measurement, which is important for real time traffic, is considered. The associated OAM standard for ATM from ITU-T is inadequate for these purposes and there is no formal approach to the topic in literature although a few schemes have been proposed. The problems of lack of schemes for accurate delay measurement, of algorithms which optimally sample for delay, and of a formal theory for evaluation of these aspects were considered here. Research directions being pursued to resolve satisfactorily the three issues mentioned above are presented in the paper.

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